PTO 03-3690

German Patent No. DE 41 26 815 A1

VOICE-DECODING SYSTEM WITH DTMF REGENERATOR AND METHOD FOR REGENERATING A DTMF SIGNAL

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UNITED STATES PATENT AND TRADEMARK OFFICE
WASHINGTON, D.C. JUNE 2003
TRANSLATED BY THE RALPH MCELROY TRANSLATION COMPANY

FEDERAL REPUBLIC OF GERMANY
GERMAN PATENT OFFICE
PATENT NO. 41 26 815 A1
(Offenlegungsschrift)

Int. Cl.[5]:	H 04 Q 1/45
Filing No.:	P 41 26 815.6
Filing Date:	August 9, 1991
Date Laid Open to Public Inspection:	February 11, 1993

VOICE-DECODING SYSTEM WITH DTMF REGENERATOR AND METHOD FOR REGENERATING A DTMF SIGNAL

[Sprachdecodierungssystem mit DTMF-Regenerator und Verfahren zur Regenerierung eines DTMF-Signals]

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Examination request according to §44 Patent Act has been filed

The invention concerns a voice-decoding system with a DTMF regenerator and a method for regenerating a DTMF signal. DTMF is the abbreviation for "dual-tone multi-frequency." The voice-decoding system and the method according to the invention use a DTMF signal that has been guided by means of a coder and a decoder. The operation of this system is nonlinear.

More specifically, the invention concerns a voice-decoding system with a DTMF regenerator for use in a digital telephone system.

DTMF signals are used in the telephone industry for transmitting command signals, especially for the remote control, e.g., of answering machines. The DTMF standard, as applied, e.g., in North America and Europe, stipulates that, in order to be valid, each pair of DTMF tones must contain one tone (row tone) from the so-called low-frequency group of 697, 770, 852, and 941 Hz and one other tone (column tone) from the so-called high-frequency group of 1209, 1336, 1477, and optionally 1633 Hz.

Voice-coding systems, especially with a converter, which operates at a low bit rate, as described, e.g., in "speech codec for the European mobile radio System" (Conference Proceedings,

ICASSP, 1988), lead to considerable distortion of the sinusoid of the DTMF signal. This effect can be traced back to the fact that the voice conversion is based on a model that is optimized only for the stated purpose. A DTMF signal that is subject to such voice conversion undergoes strong nonlinear amplitude distortion. Such distorted DTMF tones cannot be detected reliably by standard DTMF detectors, like those provided in standard terminals. Such detectors normally consider only noise and frequency distortion. Therefore, in order to use such standard DTMF detectors in standard terminals in connection with a nonlinear voice coding system, the distorted DTMF signal must be regenerated.

In voice-coded systems up to now, the DTMF detection was not performed directly on the signal passing through the standard coder/decoder circuit; instead it was handled in the framework of the signal procedure, e.g., during establishment of the connection in a European GSM system (GSM recommendation paragraph 03.14, version 3.0.2, January 1990; paragraph 04.08, version 3.12.0, March 1991).

In addition, previously there was doubt whether a standard DTMF tone with the necessary pulse length of 40 ms could be used, which is emphasized expressly in the specification for the GSM standard (GSM recommendation paragraph 06.10, Annex A 1.3.2, version 3.2.0 January 1991). Consequently, this standard requires special DTMF receivers, which had to be designed exclusively for this purpose, in order to enable DTMF communications. In order to avoid this situation, a DTMF compatible voice-decoding system is necessary.

Due to the previously mentioned difficulties, DTMF detection was not used in previous voice-coded, nonlinear systems. In linear systems of the analog or digital type, DTMF detection is known. Here, DTMF recognition is realized through simple filtering and then detection. Such techniques cannot be used for nonlinear systems. Other techniques for linear systems, such as DTMF recognition by means of known LPC (linear predictive coding), as described, e.g., by B. I. Pawate, W. Steenaart, and B. Sankur in a publication: "The DTMF Receiver Based on Linear Prediction" (Proc. of the Twelfth Biennial Symposium on Communications, Queen's University, Kingston, June 4-6, 1985), have been proposed in connection with voice coding for voice communications (US 48 53 958). However, here the DTMF signal is not distorted by the voice coding system, because the signal does not pass through the coding system at all. Instead, DTMF recognition is performed on the uncoded input of the coder, where LPC parameters that were determined by the coder are used.

Therefore, the task of the invention was to create a device and a method, which can make accessible DTMF signals that pass through a voice coding system, particularly with a low bit rate, for standard DTMF detectors.

This task is accomplished by the features of the main claim and also by the features of Claim 5. The other claims give configurations according to the invention.

According to the invention, the distorted DTMF signal is regenerated. For this purpose, a regenerator is used, which can be connected to or integrated into the voice decoding system. The DTMF regenerator corrects the distorted DTMF signals to a standard DTMF signal form. In order to enable the device containing the voice decoding system to be connected to standard DTMF compatible terminals, such as answering machines and the like, the regenerator contains a detector, which recognizes the DTMF signal. The detector includes a high-pass filter and a low-pass filter in order to separate the DTMF frequency groups and a processing unit, which determines parameters of the decoded signal, such as auto-correlation and high-band power and low-band power. These values are used to verify an incoming DTMF signal and also to calculate and verify a column tone and a row tone. In addition, these tones are then used to form and verify a number or character value, which is transmitted to the generator. In the generator, the delay resulting from the detection is corrected and a standard DTMF signal corresponding to the verified number or character value with the correct length is generated as the output signal.

The invention is explained in more detail with reference to Figures 1-5. Shown are:

Figure 1, a schematic diagram of the allocation of the decoding system according to the invention with a DTMF regenerator for the voice coding system;

Figure 2, a schematic block diagram of a configuration of the regenerator consisting of detector and generator;

Figure 3, a schematic block diagram of a first part of the method for detection according to the invention;

Figure 4, a schematic block diagram of a second part of the method for detection according to the invention;

Figure 5, a schematic block diagram of the DTMF generator according to the invention.

The DTMF regenerator consists of a DTMF detector and a generator. The detector operates according to a "sample by sample" method. Each incoming sample is fed to the detector. In the detector, after a predetermined number of L samples, a test for the presence of a DTMF signal is performed. It is required that every two L samples give the same result. The incoming samples are filtered in order to obtain a high-band group and a low-band group; for both groups, auto-correlation coefficients and power are determined. After L samples have been input, the presence of a DTMF signal is tested. For this purpose, at first the determined power values are inspected. Here, the following criteria are used:

The power in the high-band group (HBE) and in the low-band group (LBE) must be greater than a minimum value;

The LBE must be relatively stationary (compared with earlier detections);

The ratio between the LBE and HBE must lie within a certain range.

If the power criteria are satisfied, then frequency/magnitude tests are performed. Here, the term frequency/magnitude refers to the complex plane (Z plane). As soon as the criteria are not fulfilled one time for the power test or frequency/magnitude test, the frequency/magnitude criteria are switched to a larger range. This coarse range remains until both the power and also the frequency/magnitude criteria have been fulfilled for the first time.

When the frequency/magnitude criteria, which can be adapted individually for each frequency, have also been fulfilled, the number or character value corresponding to the frequencies is determined according to the DTMF standard, and the frequency/magnitude criteria are set to a narrower range (fine range) with stricter requirements.

According to Figure 1, the regenerator 15 is connected to the decoder 14. Preferably, the generator 15 forms a part of the decoder 14. Due to reasons of clearness, and not of limitation, the regenerator 15 is connected to the output device 16. The decoder 14 is fed via a transmission device 13 from the coder 12, which receives its signal from an input device 11.

According to Figure 2, the regenerator 15 contains a detector 17 and a generator 18. The detector 17 contains a low-pass filter 1 and a high-pass filter 2 in order to separate the high-frequency group of the DTMF signal from the low-frequency group. For both groups, there are devices 3 and 4 for updating the parameters (auto-correlation coefficients and power). The devices 6, 7 are used to compare the parameters, which represent a possible DTMF signal, with power and frequency/magnitude criteria and to determine a row tone and a column tone. From both values, a possible DTMF number or character (in the following called generally "character") is determined and tested relative to its correct sequence with earlier possible DTMF characters in a DTMF character validation unit 8. When a possible DTMF character within a correct sequence has been found, this information is used to control the generator 18. The generator 18 contains a DTMF generator (tone generator) 9, a delay element 5, and a multiplexer 10, which outputs a DTMF tone when a DTMF signal has been detected, otherwise it outputs the decoded signal 19.

According to a special configuration of the detector according to the invention, an algorithm is provided, which controls the generator. The input signal to the algorithm is a sample 21, which comes from the output of the decoder. The detector operates according to a sample-by-sample method, although a detection is only performed after every L samples. Detection is guaranteed by setting 31 a starting value for L and decrementing 29 by one for each sample and then comparing 30 with zero. A device 22, which either outputs the value "true" or "false," branches on one hand to the side, in which the determination of the auto-correlation coefficients and the power evaluation is performed, or alternately to the side, in which the detection of a DTMF signal is performed. The device 22 toggles between the two values for each incoming sample. This reduces the amount of input signals by a factor of 2. This also reduces the number of necessary operations.

When the device 22 has the value "true," the input signal 21 is filtered with a high-pass filter 25 and a low-pass filter 23. At each filter output, the auto-correlation coefficients are updated 24, 26. The power estimates in both ranges are likewise updated 28. Both the auto-correlation coefficients and also the power estimates are updated in a recursive method. The LBE and HBE are updated as follows:

$$LBE(n) = aLBE(n-1) + (1-a)xL(n)xL(n)$$

$$HBE(n) = aHBE(n-1) + (1-a)xH(n)xH(n)$$

where n is the time index, $a < 1$ determines how quickly LBE and HBE are updated and how large the deviation of LBE and HBE is, and xL and xH are the output signals from the low-pass filter and the high-pass filter. The auto-correlation coefficients RL(n, i) and RH(n, i) are updated in a similar way:

$$RL(n,i) = bRL(n,i) + (1-b)xL(n) * xL(n-i)\ ,\ i = 0,\ 1,\ 2$$

$$RH(n,i) = bRH(n,i) + (1-b)xH(n) * xH(n-i)\ ,\ i = 0,\ 1,\ 2.$$

In order to obtain quick power estimates and reliable auto-correlation coefficients, normally $a < b < 1$. After this update, the counter for the samples is decreased by 1 in step 29, and the algorithm returns to its original state, which means that it is ready to register the next incoming sample. When the device 22 is "false," the sample counter is tested 30. If the sample counter is not equal to zero, the algorithm returns to the start. If the sample counter is equal to zero, this value is reset 31 to the predetermined starting value L. L is determined by the minimum tone length of the DTMF signal. Every two L samples must fit completely into the minimum tone length. This requirement was set in order to make the detection less susceptible to distorted samples.

In the algorithm, five power tests 32, 33, 34, 35, 36 are performed. If LBE is smaller than a threshold T1, the algorithm continues with B. If HBE is smaller than a threshold T2, the algorithm likewise continues at point B. The power tests 32 and 33 reproduce the requirement of a lower limit on the DTMF tone amplitude. These two tests sort out all signals that are not DTMF tones with greatest likelihood. If the ratio between LBE and an earlier LBE value (LBE old) is greater than a value T3, the algorithm continues at point B. This test is performed in order to determine whether the power has suddenly fallen, which can happen at the end of a DTMF tone. If the ratio between LBE and HBE is greater than a threshold T4, the algorithm continues at point B. If the ratio between HBE and LBE is greater than a threshold T5, the algorithm continues at point B. These two tests ensure that the power difference in the high-frequency group and the low-frequency group is within certain limits. Otherwise the algorithm continues at point B.

According to Figure 4, at point B the auto-correlation coefficients are set to zero; the criteria for the frequency-magnitude test are switched to the coarse range and the possible character D(k) is set to "invalid" 49. The auto-correlation coefficients are set to zero, so that a more reliable estimation is possible for the next detection. If the algorithm continues at point A, the row

tone (high-frequency group) is calculated. The calculation is based on the auto-correlation coefficients RH(n,i). With the help of the auto-correlation coefficients, a set of derived LPC parameters is calculated. These derived LPC parameters correspond to the most dominant frequencies and magnitudes. All possible row tones have two frequency and magnitude thresholds, of which one allows large frequency and magnitude deviations (coarse values) and the other allows only small frequency and magnitude deviations (fine values). These thresholds can be set individually for each of the frequency/magnitude values to be tested. The derived LPC parameters are tested for agreement with one of the frequency and magnitude thresholds. If the thresholds are maintained, the column tone is set to a valid value, which corresponds to the column frequency. Otherwise it is set 37 to an invalid value. If the column tone is invalid, the algorithm continues at point B. The same method is used for the row tone, wherein the auto-correlation coefficients RL(n,i) for the low-frequency group are used. For this there are likewise different frequency and magnitude values (coarse values and fine values). Both can be set individually for each frequency/magnitude. If the row tone is invalid, the algorithm continues at point B. If both tones are valid, the threshold table for the frequency/magnitude testing is switched to fine values 41, which are used for the next detection, because a possible DTMF tone has been detected. Column and row tone are combined in order to form 42 a possibly valid character D(k). The final step in the detection is a character validation, which is performed both for the possibly valid character values and also for the invalid character values. The validation is performed with the three most recent possible characters D(k), D(k-1), and D(k-2). If D(k) = D(k-2), either no new character was detected 43, or two consecutive characters are not equal, and the algorithm returns to the start. If D(k) <> D(k-1), the requirement of two consecutive characters of equal value is not fulfilled, and the algorithm returns to the start. Otherwise, the value of D(k) is tested 45. If D(k) is invalid, a variable LDD (last detected digit) is set to invalid, and the variable DD (detected digit) is set 46 to invalid, and the algorithm returns to the start. If D(k) is valid, LDD is tested 47. If LDD is valid, a sequence of characters without the required pause between DTMF signals is present, and the algorithm returns to the start. If LDD is invalid, LDD is set to valid and DD is set 48 to the valid value of the character D(k), and the algorithm returns to the start. DD is only changed if a sequence of two valid and equal characters appears in combination with two invalid characters.

The algorithm ensures that if DD has been set to invalid before the algorithm is called, this only outputs a valid value DD if a new DTMF tone is detected only once. If DD is not changed from call to call of the algorithm, DD is output as long as the DTMF signal is detected. The latter solution is preferably used in connection with the DTMF regenerator.

According to Figure 5, the inputs of the DTMF generator 18 are the value DD and the recorded samples x(n). Because the detector performs a detection after only every L samples and requires two consecutive detections to form a valid character, there is an inherent delay caused by

the detection. The theoretical maximum value of this delay is 3L, although the actual maximum delay is less than 3L but greater than 2L. The DTMF generator 18 must compensate for this delay so that voice-distorted DTMF tones are not missed by a connected device. The DTMF generator consists of a delay element 50, a tone generator 51, and a multiplexer 52. The size of the delay element is 3L samples corresponding to the theoretical maximum delay. The input parameter for the delay element is the output signal of the decoder x(n), and the output parameter of the delay element is x(n-3L), which is relatively speaking an old output signal from the decoder. As long as DD is invalid, the multiplexer 52 selects the output from the delay element. As long as DD is valid, the multiplexer selects the output from the tone generator 51. The tone generator generates the DTMF tone, which corresponds to the value DD. In this way, no voice-coded DTMF tone is missed.

Claims

1. Voice-decoding system, characterized in that there are devices (15) for regenerating a standard DTMF signal from an input signal, which contains a distorted DTMF signal.

2. Voice-decoding system according to Claim 1, characterized in that the devices (15) for regenerating a standard DTMF signal contain devices for detecting (17) a DTMF signal and devices for generating (18) a standard DTMF signal.

3. Voice-decoding system according to Claim 2, characterized in that the devices (18) for generating a standard DTMF signal contain a tone generator (51), which is connected to the devices (17) for detecting a DTMF signal, and also a delay element (50), which is connected to the input signal, as well as a multiplexer (52), which is connected to the delay element (50), to the tone generator (51), and to the devices for detecting a DTMF signal (17).

4. Voice-decoding system according to Claim 2 or 3, characterized in that the devices for detecting (17) a DTMF signal from the input signal contain devices (1,2), which can separate the input signal into a high-frequency group and a low-frequency group, devices for determining auto-correlation coefficients and the power (3,4) for both frequency groups, devices for comparing the power of both groups with predetermined thresholds, devices for comparing the frequency groups with set frequency and magnitude criteria, devices for calculating a character (6, 7) from both frequency groups, and also devices (8) for comparing this character with earlier characters.

5. Method for regenerating a standard DTMF signal from an input signal, which contains a distorted DTMF signal and which is used to control a generator, characterized in that the distorted DTMF signal is detected, the detected DTMF signal is tested with earlier detected DTMF signals, a detected character DD is formed for the presence of a correct sequence of detected DTMF signals, a DTMF generator is controlled with the detected character in order to form a standard DTMF

signal, where the output of this DTMF generator and the input signal are input into a multiplexer and the output of the multiplexer is controlled by the detected character DD.

6. Method according to Claim 5, characterized in that the digital output of a decoder is used as the input signal (21), every second digital output signal of the decoder is taken as a sample for detecting the distorted DTMF signal (22), these samples are filtered (23,25) in order to separate a high-frequency group of samples from a low-frequency group of samples, auto-correlation coefficients (24,26) and power estimates (28) are updated recursively for both frequency groups, the samples are counted (29), after a certain number of samples the power estimates are tested for agreement with predetermined criteria (32,33,34,35,36), if the power criteria are satisfied, frequency and magnitude values are determined from the auto-correlation coefficients (37), these frequency and magnitude values are tested for agreement with predetermined criteria, if these criteria are satisfied, a possibly valid character (37,39), otherwise a possibly invalid character (49), is formed, testing of this character for a required sequence according to earlier characters of the same value [occurs], [and] testing of consecutive characters of the same value (43,44) for combination of two consecutive valid characters with two consecutive invalid characters (45,46,47), [occurs] wherein, if this combination is found, a detected character DD is set (48) and the system waits for a new sample.

7. Method according to Claim 6, characterized in that the auto-correlation coefficients are set to zero (49), if the power or frequency/magnitude criteria are not fulfilled.

8. Method according to Claim 6 or 7, characterized in that the frequency/magnitude tests (37,39) are performed with individual criteria for each frequency and magnitude.

9. Method according to Claim 6 or 8, characterized in that after determination of a possibly valid character, the frequency and magnitude criteria are set to a fine value (41).

10. Method according to Claim 6 or 9, characterized in that the test for the correct sequence of possibly valid or possibly invalid characters is applied to the three most recent characters D(k), D(k-1), D(k-2) (43, 44, 45).

11. Method according to Claim 10, characterized in that no detected character is determined if D(k-2) = D(k); no detected character is determined if D(k-1) does not equal D(k); and no detected character is determined if D(k) is also invalid.

12. Method according to Claim 11, characterized in that for a valid character D(k), a variable LDD is set to valid and a variable DD is set to valid (48) if LDD was invalid (47).

13. Method according to Claim 12, characterized in that DD is transmitted to a tone generator (51) and a multiplexer (52), the samples are transmitted to a delay element (50), which has a delay of three times the magnitude of the determined number of samples, the output of the multiplexer (52) is controlled by DD, wherein the samples are output as long as DD is invalid, otherwise the signal is output by the tone generator (51).

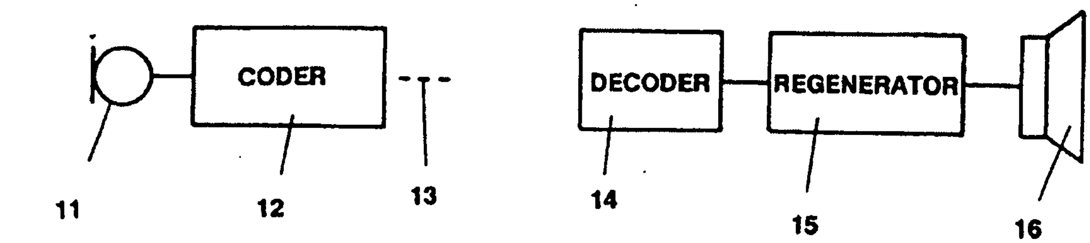

CODER
DECODER
REGENERATOR
11
12
13
14
15
16

Fig. 1

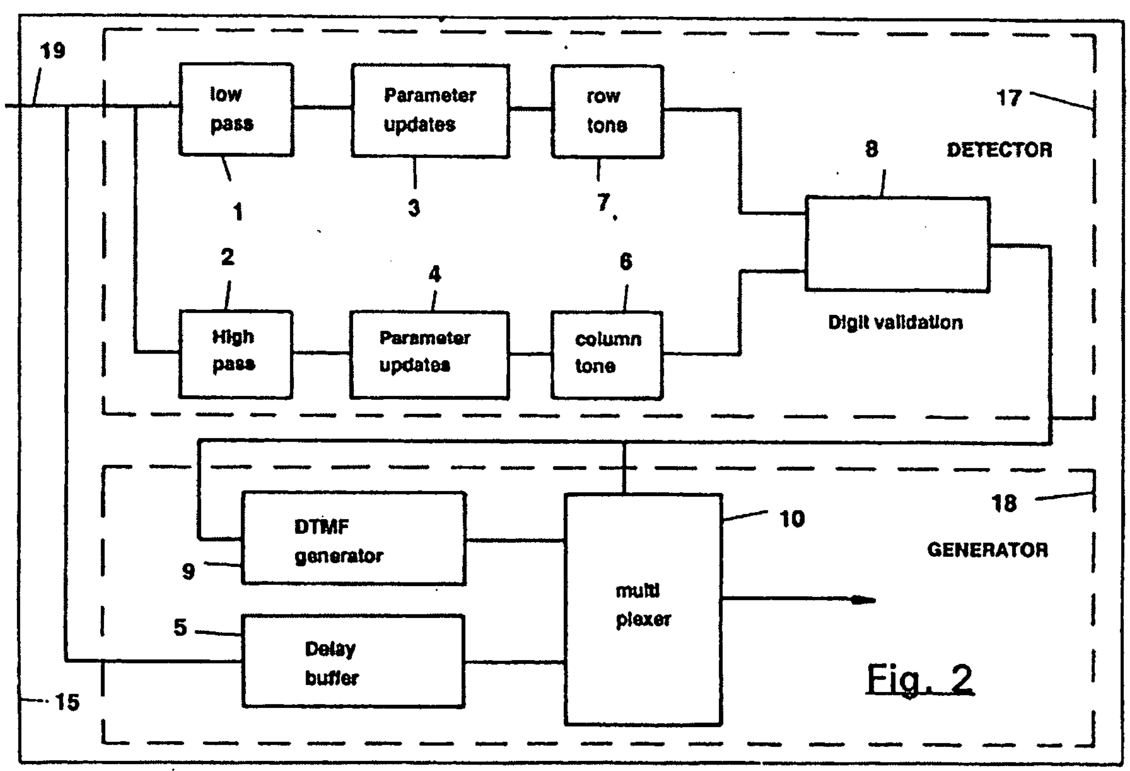

19
low pass
Parameter updates
row tone
17
DETECTOR
8
1
3
7
2
4
6
High pass
Parameter updates
column tone
Digit validation
DTMF generator
9
10
18
GENERATOR
multi plexer
5
Delay buffer
15

Fig. 2

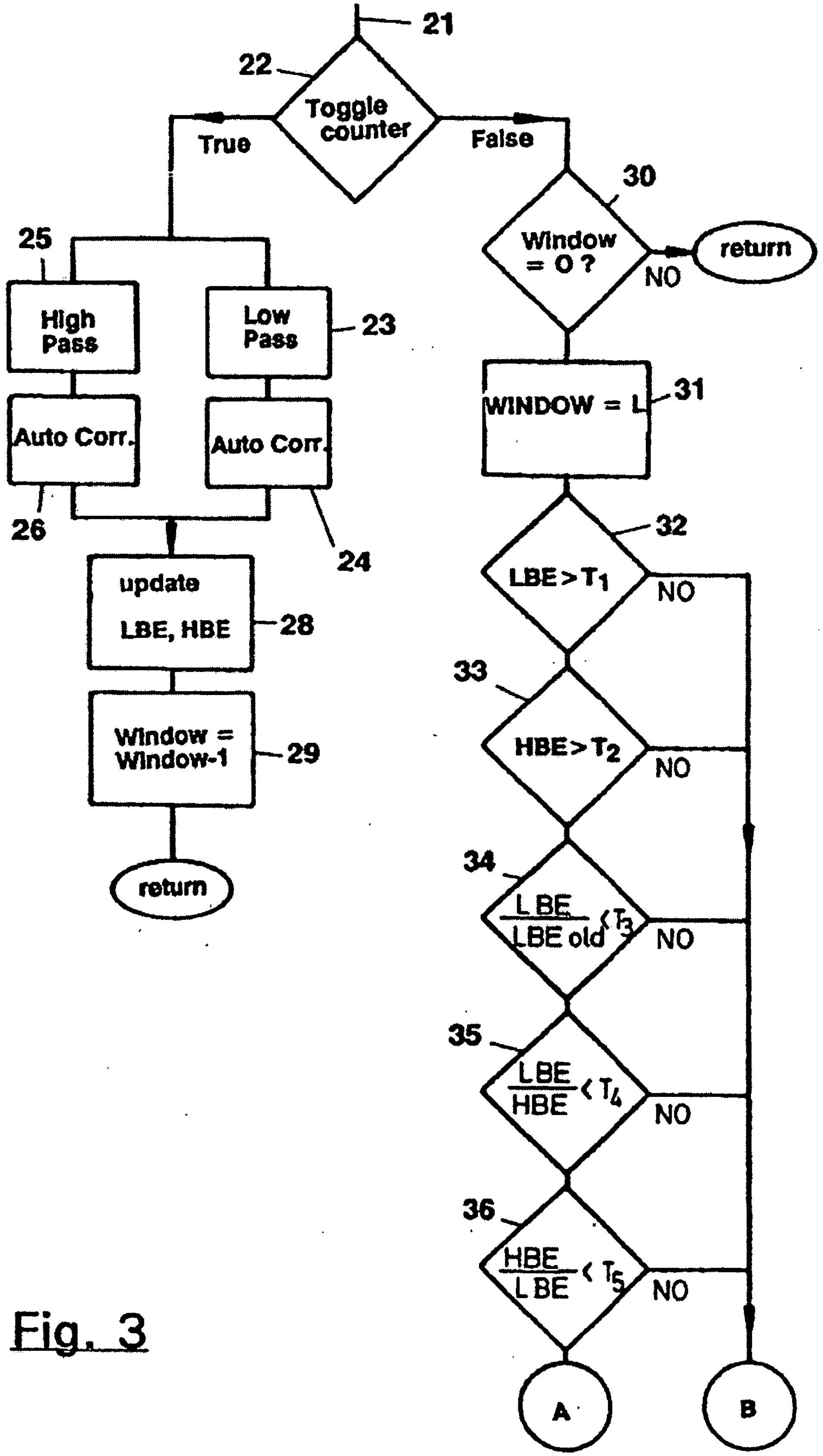

21
22
Toggle counter
True
False
30
Window = 0 ?
NO
return
25
High Pass
Low Pass
23
Auto Corr.
Auto Corr.
WINDOW = L
31
26
24
32
LBE > T1
NO
update LBE, HBE
28
33
HBE > T2
NO
Window = Window-1
29
return
34
LBE / LBE old < T3
NO
35
LBE / HBE < T4
NO
36
HBE / LBE < T5
NO
A
B

Fig. 3

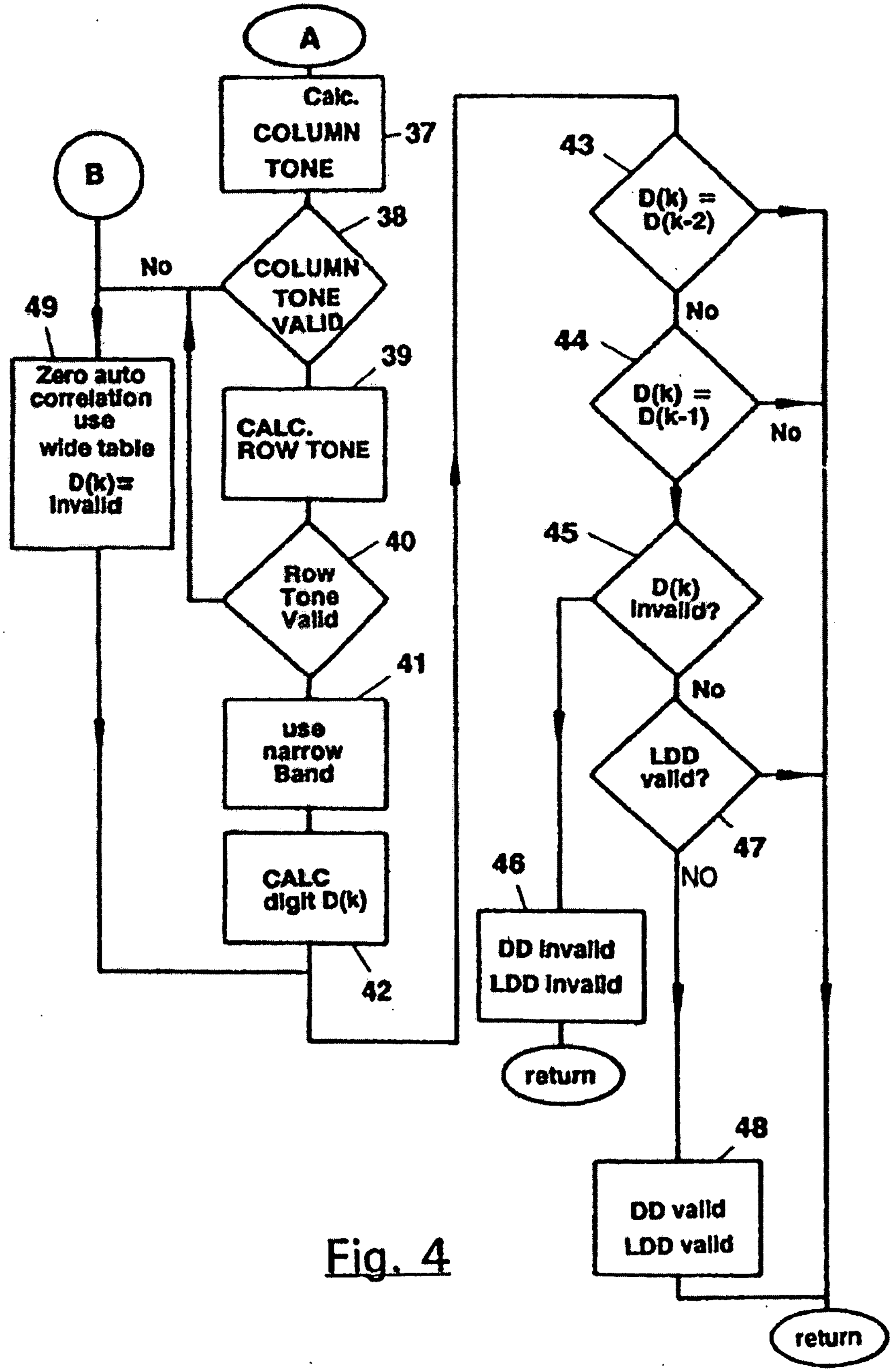
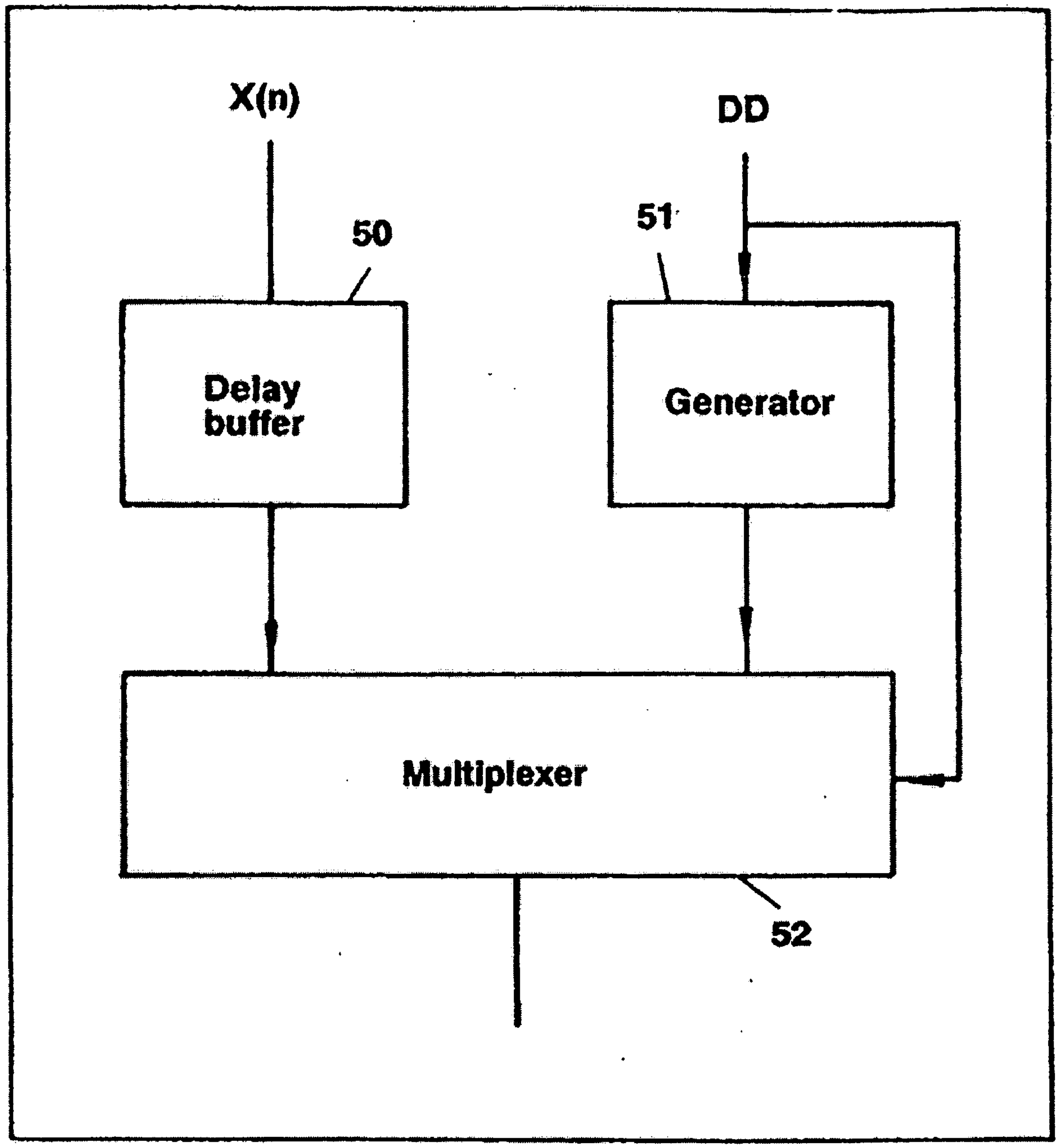

A
Calc.
COLUMN
TONE
37
B
38
No
COLUMN
TONE
VALID
49
Zero auto
correlation
use
wide table
D(k)=
Invalid
39
CALC.
ROW TONE
40
Row
Tone
Valid
41
use
narrow
Band
CALC
digit D(k)
42
43
D(k) =
D(k-2)
No
44
D(k) =
D(k-1)
No
45
D(k)
Invalid?
No
LDD
valid?
47
NO
46
DD invalid
LDD invalid
return
48
DD valid
LDD valid
return

Fig. 4



DTMF generator

Fig. 5